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KAZEMINEZHAD, FARZAD	

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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

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Office Action Summary	**Application No.** 10/579,235	**Applicant(s)** NAKADAI ET AL.
	Examiner FARZAD KAZEMINEZHAD	**Art Unit** 2626

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☐ Responsive to communication(s) filed on _____.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) 1-12 is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) 1-12 is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on _____ is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☒ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☒ All b)☐ Some * c)☐ None of:

1.☒ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☒ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date 1/11/2010.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application
6) ☐ Other: _____.

DETAILED ACTION

Continued Examination Under 37 CFR 1.114

1. A request for continued examination under 37 CFR 1.114, including the fee set forth in 37 CFR 1.17(e), was filed in this application after final rejection. Since this application is eligible for continued examination under 37 CFR 1.114, and the fee set forth in 37 CFR 1.17(e) has been timely paid, the finality of the previous Office action has been withdrawn pursuant to 37 CFR 1.114. Applicant's submission filed on 3/23/2010 has been entered.

Response to Amendment

2. In response to the office action from 12/23/2009, the applicant has submitted an amendment, filed 3/23/2010, amending the independent claim 1, while arguing to traverse the prior art rejection. Applicant's arguments have been fully considered, however they are moot due to new grounds of rejection further in view of
Kim et a. (US 2004/0175006).

Response to Arguments

3. The examiner will address Applicant's Remarks in chronological order although the examiner ties common arguments together.

Pages 10 and 11 provide general overview of the last office action, the references used and the first claim. On page 12 the first ¶, the last four lines it is asserted that: "Asano (US 2004/0054531) does not teach or suggest that a sound

direction is localized .... based on the acoustic signals detected by the plurality of microphones,…., Rather, as clearly indicated in Asano, the direction of the sound is estimated from a power difference of phase difference". This argument is flawed for several reasons: 1) the power and phase difference is due to the signals (acoustic signals) detected by the two (plural number of microphones) of the robot as mentioned on page 6 of the last action; 2) The applicant is respectfully questioned: don't power and phase correspond to acoustic features which are imperatively obtained by detection of acoustic signals which as mentioned are detected by two microphones (module 21 Fig. 2, 9 and 11 of Asano) on page 6 of the last action; furthermore power is maximized if either the beam of the array of microphones is steered physically turning its microphones or by simply adjusting delays/phase differences toward sound source producing the signals; last but not least even the application utilizes phase and "intensity" difference in its direction localization as page 5 the second ¶ states: "It may be possible that the sound source localization module employs scattering theory... to specify sound direction for the speaker with the intensity difference and the phase difference detected through the plurality of microphones". Energy, power, spectral characteristics including the phase do correspond to acoustic features as for instance US 2008/0126100 ¶ 0048 lists and this is well known in the art. On page 12 the 2nd ¶ last 3 lines a similar argument is made.

The examiner will next address a point made countless number of times by the Applicant throughout the remarks which appears to be fundamental to either the novelty of his invention and or what distinguishes the application from Asano, namely why

Asano does not teach or render obvious storing direction-dependent acoustic models that are adjusted to a plurality of directions at intervals (page 13 the 1st ¶ lines 3-4 and the 2nd ¶ last 4 lines, page 14 1st ¶ lines 1-3 and the second ¶ lines 7-5 above the 3rd ¶, page 15 the 1st ¶ lines 4-8 and lines 5-1 above the 2nd ¶, page 16 the 1st ¶ lines 3-8). On page 17 the 2nd ¶ lines 6-8 it is further pointed out that: "mere allegations that "direction-dependent acoustic models" phrase in the last limitation of claim 1 is to be replaced with "distance-dependent acoustic models" and ¶ 114 is not sufficient".

The new amended limitation though has used a more definite language in how acoustic model for a given sound direction is composed by dropping the word "adjust" for which therefore a new reference was used, however the examiner will provide more information on this subject to clarify how the new office action maps the new amended limitation of claim 1. In the new office action in response to the amendment to claim 1, a new reference Kim et al. (US 2004/0175006) which teaches generating direction dependent acoustic models is used and therefore since it was no longer rendered obvious replacing distance-dependent with direction dependent acoustic models using the single reference Asano, therefore the above arguments are no longer valid and the applicant is respectfully directed to the office action that follows for more details. The short answer to the above remarks in light of the new reference is that, by incorporating the functions of the module responsible for enabling Kim et al.'s acoustic models becoming direction dependent into Asano, Asano's acoustic models are suggested to become direction dependent as well. Graham versus Deere § D [3] states: " a finding that one of ordinary skill in the art would have recognized that applying the known

technique would have yielded predictable results and resulted in an improved system”. Therefore, suggesting that Asano improves his model by adjusting his distance dependent acoustic models to direction falls well within this guideline since Asano also teaches obtaining the direction of sound. Note that the specification hardly explains what exactly constitute an acoustic model which is direction dependent, or how a direction dependent acoustic model differs a regular acoustic model. The best explanation that could be found is based on the teachings of page 29 the second ¶ lines 3-6, which appear to teach direction dependent acoustic models are obtained by generating or extracting direction dependent acoustic features (e.g. spectral attributes such as frequency). Kim et al. also teaches obtaining direction dependent peaks each of which are associated with given frequencies.

On page 13 the first ¶ the last 3 lines it is asserted: "Asano specifically describes that a set of acoustic models for a distance is stored”. This statement hides the fact that Asano also stores acoustic models for different distances: i.e., ¶ 0114 lines 4-5 teach: "acoustic modes of speeches …. are produced while varying distances”. Indeed this is also quoted by the Applicant on page 14 last ¶. Likewise on page 15 the first ¶, line 5 above the second ¶, it is asserted that: “In Asano, the acoustic models are selected rather than being composed". By the same quotation above, it should be clear that in Asano acoustic models are produced (composed) and not just "selected”.

On page 16 the last line and page 17 the first line, it is asserted: "No evidence was provided in the Office Action to support such (i.e., obtaining direction dependent acoustic models by using stored distance-dependent acoustic models using the

methods of 129 ... stated in the obviousness of claim 1) holding". A similar remark was also made on page 17 the second ¶ and the last ¶, and in the last ¶ of page 17 MPEP 2144.03 was quoted and the examiner was educated that "the examiner should cite a reference in support of his or her position". Similar remarks were also made on page 18 the second ¶ and the third ¶ where it was stated : "This is another indication that rather than finding a single reference that describes all the features recited in the independent claims, the examiner has opted to make his own modifications of the description of Asano ...". All these arguments appear to be focused on the rationale of using a single reference obviousness motivation for the first claim based on the reasoning above. These arguments are no longer relevant because now a second reference which specifically addresses obtaining direction dependent acoustic features and models is used.

On page 19 the first ¶, the Applicant has asserted claims 4, 6-7 to be allowable by virtue of their dependence on the allegedly patentable claim 1 without discussing their own allowable subject matters and are therefore moot.

On page 19, the third ¶ and page 20 the first ¶ the Applicant has simply provided an overview of the claim 2 and contended it to be allowable because it has “similar features as those recited in independent claim 1” (page 19 last line) and are therefore moot.

From the 3rd ¶ on page 20 to the first ¶ on page 22, the Applicant has contended that Ito (7,076,433) and Okuno et al. (7,035,418) fail to teach other dependent claims (e.g. 2, 3, 8-12 and 5) because they fail to teach the limitations that they were not used

for! or basically they fail to treat the deficiencies of Asano!! Here the Applicant is reminded that: FP-07-37-13 teaches:

In response to applicant's arguments against the references individually, one cannot show nonobviousness by attacking references individually where the rejections are based on combinations of references. See *In re Keller*, 642 F.2d 413, 208 USPQ 871 (CCPA 1981); *In re Merck & Co.,* 800 F.2d 1091, 231 USPQ 375 (Fed. Cir. 1986).

The Applicant is next respectfully directed to the office action that follows.

Claim Rejections - 35 USC § 103

4. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

5. Claims 1, 4, 6-7 are rejected under 35 U.S.C. 103(a) as being unpatentable over Asano (US 2004/0054531), and further in view of Kim et al. (US 2004/0175006).

Regarding claim 1, Asano does teach an automatic speech recognition system, which recognizes speeches in acoustic signals detected by a plurality of microphones as character information, the system comprising:

a sound source localization module configured to localize a sound direction corresponding to a specified speaker based on the acoustic signals detected by the plurality of microphones (¶ 0129 teach the head unit 3 in Fig. 2 (performing similar function as the sound source localization module) enables obtaining the direction of sound utilizing a plurality of microphones at a target (e.g. a robot) which receives the speech signals by computing power and phase differences of speech signals due to sources attributed to users (speakers); Abstract teaches all incoming speech undergo speech recognition utilizing plural sets of acoustic models);

a feature extractor configured to extract features of speech signals contained in one or more pieces of information detected by the plurality of microphones (the feature extractor unit 101 in Fig. 9 receives speech data from microphones (unit 21 Fig. 9) via the analog to digital converter. ¶ 0104 lines 1-4 teach extracting feature vectors of the incoming speech data) ;

an acoustic model memory configured to store distance-dependent acoustic models that are adjusted to a plurality of distances at intervals (¶ 0131 lines 4-8 referring to Fig. 9 disclose storing acoustic models corresponding to selected distances in the database units (104)_1 …. (104)_N (located in the memory module 42 in Fig. 3); these data are used following determination of distance of a sound source attributed to a user; ¶ 0106 teaches all feature vectors corresponding to acoustic analysis (¶0104 lines 1-2) are stored in speech periods (intervals) corresponding to an utterance);

an acoustic model composition module configured to compose an acoustic model adjusted to the sound distance, which is localized by the sound source localization

module, based on the distance-dependent acoustic models in the acoustic model memory, the acoustic model composition module also configured to store the acoustic model in the acoustic model memory (¶ 0114 teaches "N" acoustic models corresponding to “N” sound sources located at “N” distances are produced (composed) where each acoustic model corresponding to a certain distance (localization) is stored in a certain database (e.g. one of the units (104)_1 to (104)_N) in Fig. 9));

and a speech recognition module configured to recognize the features extracted by the feature extractor as character information using the acoustic model composed by the acoustic model composition module (¶ 0132 (lines 2-1 above ¶ 0133) teaches module units 41B and 41A in Fig. 3 as disclosed in step S4 in Fig. 10 enable performing speech recognition utilizing feature vectors extracted from the speech data (¶ 0132 lines 1-3); ¶ 0108 lines 1-4 teach acoustic model databases used for the speech recognition have stored acoustic characteristics of phonetic-linguistic-units such as phonemes or syllables (e.g. character units comprising words) in them).

Asano does not specifically disclose:

<u>Wherein the acoustic model composition module is configured to compose an acoustic model for the sound direction.</u>

Kim et al. does teach an <u>acoustic model composition module configured to compose an acoustic model for the sound direction</u> (The acoustic model (generated by the voice recognizer (¶ 0009 lines 1-3)) used by the voice recognizer (¶ 0053 lines 8-6 above ¶ 0054) receives a voice signal which has had an acoustic source direction detection performed on it by the peak detection module unit 461 in Fig. 4 (direction

dependent acoustic model composition unit) as disclosed in ¶ 0016 last 8 lines and ¶ 0052 by determining a direction associated with a given frequency (acoustic feature) peak).

It would have therefore been obvious to one with ordinary skill in the art at the time the invention was made to incorporate module 461 in Fig. 4 of Kim et al. into the feature extractor module 101 of Fig. 9 of Asano would enable the combined modules to function in combination as they do separately and to further enable Asano to not only generate and store distance-dependent acoustic models in the memory as is done here in modules 104 in Fig. 9, but also direction-dependent acoustic models obtained by using the newly determined direction dependent acoustic features so that the robot will have bias for direction as well as distance and will point at the direction of the source associated with the sound frequency to reduce detecting wrong signals and enhance its speech recognition performance.

Regarding claim 4, Asano does teach a system according to claim 1, wherein the sound source localization module is further configured to employ a scattering theory <u>to</u> generate a model for an acoustic signal, which scatters on a surface of a member (¶ 0142 lines 4-9 and ¶ 0145 teach using reflected (scattered) ultrasonic wave pulses from an obstacle, the robot can determine the distance of a user producing speech from the robot which according to ¶ 0131 is used in generating distance dependent acoustic models; all these processes are made possible by the sensor unit 111 (Fig. 11) attached to the robot);

specifying the sound direction for the speaker with the intensity difference and the phase difference detected from the plurality of microphones (¶ 0129 lines 1-5);

Asano does not specifically teach using scattered (reflected) waves from the surface to which the microphones are attached to determine the sound direction attributed to a user (speaker). But it would have been obvious to one with ordinary skill in the art at the time the invention was made to utilize the sensor unit 111 to create a second model in which one determines the phase and power difference of the sound waves reflected (scattered) from the surface of the robot in determining the direction of the speech signals and thereby create a second model of determination of direction of a sound wave incident on the robot by analyzing reflected rather than incident waves on the robot and thereby help in generating more accurate sound direction by benchmarking the results of the two models against one another.

Regarding claim 6, Asano does not specifically disclose a system according to claim 1, wherein the acoustic model composition module is configured to compose an acoustic model for the sound direction by applying weighted linear summation to the direction- dependent acoustic models in the acoustic model memory, and weights introduced into the linear summation are determined by training.

Asano however does teach using N stored distance dependent acoustic models (104 units in Fig. 9) to select one acoustic model among them which has the closest match for a sound source which is located within the range of the distances corresponding to the said acoustic models by using its matching unit 103 which does

the matching by calculating acoustic scores (¶ 0122). ¶ 0121 lines 1-4 teaches distance calculator (unit 47 in Fig. 3) enabling calculation of distance of the robot from a user uttering speech (speaker).

It would have therefore been obvious to one with ordinary skill in the art at the time the invention was made to compose an acoustic model corresponding to a sound source using a linear interpolation of acoustic models which correspond to the distances closest to the said sound source and train the acoustic model by changing the coefficients of the linear interpolation to achieve the best score, and thereby compose a more realistic acoustic model for the said sound source resulting in better performance by the robot.

For extension of this discussion from distance dependent to direction dependent acoustic models please see the obviousness of the claim 1.

Regarding claim 7, Asano does teach a system according to claim 1, further comprising a speaker identification module, wherein the acoustic model memory is further configured to possess the direction-dependent acoustic models for respective speakers (identical to claim 1, 3rd limitation and rejected under similar rationale),

and wherein the acoustic model composition module is further configured to:

refer to direction-dependent acoustic models of a speaker who is identified by the speaker identifying module and to a sound direction localized by the sound source localization module (¶ 0116 teaches the acoustic model database to have the acoustic model of speakers (specific speakers) at specified locations; ¶ 0159 teaches those

acoustic models are produced by from the speech data acquired by microphone placed close to the mouth of the speakers for better voice recognition leading to better speaker identification; ¶0130 lines 5-1 above ¶0131 teach image of the face of a user (e.g. a potential speaker) taken by the robot's CCD cameras (modules 22L and 22R in Fig. 2) are used as a reference pattern (stored) for image recognition (user identification); finally ¶ 0145 lines 1-5 teach the robot uses both detection of a user uttering (identifying a user by his voice) as well as image recognition (identifying a user by his image) in orienting his head unit in the direction of the user; i.e., speaker identification is done by both image pattern recognition as well as by voice recognition using the stored acoustic models of the user (speaker));

compose an acoustic model for the sound direction based on the direction-dependent acoustic models in the acoustic model memory; and storing the acoustic model in the acoustic model memory (corresponds to the third limitation of the first claim and rejected under similar rationale).

For obviousness analysis please see claim 1.

2. Claims 2-3, 8-12 are rejected under 35 U.S.C. 103(a) as being unpatentable over Asano in view of Kim et al., and further in view of Ito et al. (US Patent 7,076,433).

Regarding claim 2, Asano in view of Kim et al. do teach an automatic speech recognition system, which recognizes speeches of a specified speaker in acoustic

signals detected by a plurality of microphones as character information, the system comprising:

a sound source localization module configured to localize a sound direction corresponding to the specified speaker based on the acoustic signals detected by the plurality of microphones (identical to the first limitation of claim 1 and rejected under similar rationale);

an acoustic model memory configured to store distance-dependent acoustic models that are adjusted to a plurality of directions at intervals (identical to the 3rd limitation of claim 1 and rejected under similar rationale) ;

an acoustic model composition module configured to compose an acoustic model adjusted to the sound direction, which is localized by the sound source localization module, based on the distance-dependent acoustic models in the acoustic model memory, the acoustic model composition module storing the acoustic model in the acoustic model memory (identical to the 4th limitation of claim 1 and rejected under similar rationale) ;

and a speech recognition module configured to recognize the features extracted by the feature extractor as character information using the acoustic model composed by the acoustic model composition module (identical to the 5th limitation of claim 1 and rejected under similar rationale)

Asano in view of Kim et al. do not specifically disclose a sound source separation module which separates speech signals of the specified speaker from the acoustic signals based on the sound direction localized by the sound source localization module

a feature extractor configured to extract features of the speech signals separated by the sound source separation module;

an acoustic model memory configured to store direction-dependent acoustic models that are adjusted to a plurality of directions at intervals corresponding to speech signals.

Ito et al. does teach a sound source separation apparatus which in one embodiment separates sound (e.g. attributed to a speaker) from a mixed input signal (acoustic signal) by utilizing sound source direction as an acoustic feature (Abstract lines 1-2 first ¶ and line 4 from the bottom; Col. 18 lines 61 to 66 referring to Fig. 16 unit 911). Furthermore it teaches its acoustic feature extractor to incorporate a sound source direction prediction layer which aids in determination of the peaks corresponding to acoustic features of the sound source incident from a certain direction (Col. 19 lines 33-40 and module 921 in Fig. 16).

It would have therefore been obvious to one with ordinary skill in the art at the time the invention was made that utilizing the modules 915 and 921 in Fig. 16 of Ito et al. into the feature extractor unit 101 of Fig. 9 of Asano would enable Asano in view of Kim et al. to separate a sound (e.g a speech signal) from a mixed input when the sound is incident from a certain direction by utilizing its direction dependent features enabling the robot to obtain bias for direction for a certain speaker and will point at the direction of the source of sound to reduce detecting wrong signals and enhance its speech recognition performance. Storage of direction dependent features also aids in avoiding calculations of the said features and helps in raising efficiency.

Regarding claim 3, Asano in view of Kim et al. do suggest a system according to claim 1, wherein the sound source localization module is further configured to:

acquire an intensity difference and a phase difference for the harmonic relationships extracted through the plurality of microphones (¶ 0129 lines 1-5 teach acquiring power (intensity) and phase difference of speech signals (which maintain harmonic spectra) picked up by a device (e.g. robot) microphones);

acquire belief factors for a sound direction based on the intensity difference and the phase difference, respectively; and

determine a most probable sound direction (utilizing power and phase difference in determining the direction of sound as disclosed in ¶ 0129 lines 1-5 does inherently involve these or equivalent steps to achieve the same net outcome (sound direction determination)).

Asano does not specifically disclose:

To perform a frequency analysis for the acoustic signals detected by the microphones to extract harmonic relationships.

Ito et al. does disclose performing a frequency analysis for the acoustic signals detected by the microphones to extract harmonic relationships (Col. 12 lines 45-50 teach harmonic calculation layer (named as the intermediate feature extraction layer unit 107 in Fig. 9) determines harmonic features of features (attributed to acoustic analysis of speech signals) at each time based on their frequency variation rates (i.e. by doing frequency analysis));

It would have therefore been obvious to one with ordinary skill in the art at the time the invention was made that utilizing the harmonic calculation layer (unit 107 in Fig. 9) into the feature extractor unit 101 of Fig. 9 of Asano would enable Asano in view of Kim et al. to extract harmonic structures attributed to speech signals incident on the microphones of the robot prior to obtaining their phase and power differences for the sake of determining the direction of the speech and thereby eliminate redundant parts of the spectra in determining direction of speech which primarily included harmonics and thereby enhance efficiency and accuracy.

Regarding claim 8, the preamble, the 1^{st}, 4^{th}, 5^{th}, 6^{th} and 7^{th} limitations correspond to the preamble, 1^{st} , 2^{nd}, 3^{rd}, 4^{th} and 5^{th} limitations respectively of the claim 1 and are therefore rejected under similar rationale over Asano in view of Kim et al..

Limitation 3, corresponds to the limitation 2 of the claim 2 and is therefore rejected under similar rationale by Ito et al.

Regarding the 2^{nd} limitation, storing direction dependent acoustic models at time intervals (the 3^{rd} limitation of claim 1) will amount to storing the sound direction at time intervals corresponding to the speech attributed to a sound source and the storage medium used (i.e. the memory unit 42 in Fig. 3) will enable the unit the functionality of the stream tracking module claimed in this limitation which will enable estimating the current position of a sound source by simple interpolation.

For obviousness please see claims 1 and 2.

Regarding claims 9, 10, 11, 12, they correspond to claims 3, 4, 6, 7 respectively with identical limitations and are therefore rejected under similar rationales.

6. Claim 5 rejected under 35 U.S.C. 103(a) as being unpatentable over Asano in view of Kim et al. and Ito et al., and further in view of Okuno et al. (US Patent 7,035,418).

Regarding claim 5, Asano in view of Kim et al. and Ito et al. do not specifically disclose a system according to claim 2, wherein the sound source separation module is further configured to employ an active direction-pass filter so as to separate speeches, the filter is configured to:

separate speeches by a narrower directional band when a sound direction, which is localized by the sound source localization module, lies close to a front, which is defined by an arrangement of the plurality of microphones;

and separate speeches by a wider directional band when the sound direction lies apart from the front.

Okuno et al. does teach a directional filter for sound based on the position of the source, enabling localization and extracting sound (speech) information of the said sound source which will thereby enable it to separate that source from other sound (speech) sources (Col. 4 lines 25-34); Col. 8 lines 1-7 referring to the flow chart on Fig. 8 identifies steps ST5 and ST6 as the steps associated with the directional filter's operation; Col. 8 lines 15-20 note the results of directional filter on detecting directions of sound from three sources (A, B and C on Fig. 7) and notes that the angular range

(directional band) about which these directions are determined to have been reduced due to the application of the filter; Col. 11 lines 48-52 teach the directional filter functions according to the direction and position of the sound source which is determined first by computing the difference between phase and intensity of sound received at two receivers from the same source which is directly related to their path differences; Col. 6 lines 32-35 referring to Fig. 4 explicitly show a relationship (d=D sin(theta)) between the two "sound" signal path differences (attributed to the source position which is called “d”), the distance between the receiving microphones (called “D”) and the angle governing the direction of the sound source with respect to an axis at right angles to the line connecting the two microphones (called “theta”) which shows the angle and thereby the angular range (directional band) to increase with that path difference.

It would have therefore been obvious to one with ordinary skill in the art at the time the invention was made that utilizing these methods (i.e., steps ST5 and ST6 in Fig. 8 of Okuno et al.) for directional filter into the flow chart of Fig. 10 of Asano (to after step S2) would enable Asano in view of Kim et al. and Ito et al. to apply the distance dependent directional filter of Okuno et al. which would result in less accuracy and larger angular range (wider directional band) for further sound sources since the incoming signals possess longer path differences as larger path difference (i.e. larger “d” in the relationship above) lead to larger angular range (i.e. larger deviations (directional band) in “theta”). Application of directional filter will reduce the error in determining the direction of the sound source in general.

Conclusion

The prior art made of record and not relied upon is considered pertinent to applicant's disclosure. Maekawa et al. (US Patent 6,471,420), Almstrand et al. (US 2003/0229495).

Any inquiry concerning this communication or earlier communications from the examiner should be directed to FARZAD KAZEMINEZHAD whose telephone number is (571)270-5860. The examiner can normally be reached on M-F 8:30AM-5:00 PM EST.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Talivaldis I. Smits can be reached on (571)272-7628. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

/FK/

/Talivaldis Ivars Smits/
Primary Examiner, Art Unit 2626 10/11/2010